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IN THE CLAIMS:

This listing of claims will replace all prior versions, and listings, of claims in the application:

LISTING OF CLAIMS:

1.   1.   (original) A method, comprising the steps of:
2.   (a) obtaining information relevant to the quality of service of voice calls being
3. transmitted from a first location to a second location via an IP network;
4.   (b) calculating a parameter based on said information; and
5.   (c) accepting a new call into the IP network in the case of said parameter not
6. exceeding an upper threshold.

1.   2.   (original) The method of claim 1 wherein said new call is accepted into the IP
2. network at a reduced bandwidth in the case of said parameter exceeding a lower
3. threshold.

1.   3.   (original) The method of claim 1 where said new call is not accepted into the
2. IP network in the case of said parameter exceeding the upper threshold.

1.   4.   (currently amended) The method of claim 1 wherein the information obtained
2. is a number of ~~lost packets, late packets and received packets (collectively defined~~
3. ~~as "sent" packets)~~ sent packets transmitted from said first location to said second
4. location in the IP network, wherein the number of sent packets comprises a number
5. of lost packets, a number of late packets and a number of received packets.

1.   5.   (original) The method of claim 1 wherein the information obtained is a delay
2. of received packets transmitted from said first location to said second location in the
3. IP network.

1.   6.   (original) The method of claim 1 wherein the information obtained is a delay
2. variation of received packets transmitted from said first location to said second
3. location in the IP network.

1.   7.   (original) The method of claim 1 wherein the information is obtained on a
2. periodic basis.

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1　8. (original) The method of claim 1 wherein the information is obtained on an
2　exception basis using an immediate report.

1　9. (original) The method of claim 1 wherein the parameter is identified as a packet
2　lost ratio (PLR).

1　10. (original) The method of claim 9 wherein PLR is defined as

2

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})} .$$

1　11. (original) The method of claim 2 wherein bandwidth is reduced for a newly
2　accepted call by selecting a first encoder to encode the new voice call information in a
3　bandwidth that is smaller than bandwidths of other calls accepted in the network that
4　are encoded by a second encoder.

1　12. (currently amended) The method of claim 2 wherein the bandwidth of a newly
2　accepted call is reduced by increasing the packet size (voice sample) for said newly
3　accepted voice call, wherein the packet size is indicative of a size of a corresponding
4　voice sample.

1　13. (original) The method of claim 2 wherein the bandwidth of a newly accepted call
2　is reduced by activating the characteristic of silence suppression for said newly
3　accepted voice call.

1　14. (original) Apparatus comprising a gateway for interfacing voice call data from a
2　public switch telephone network to an internet protocol network; said gateway further
3　comprising:
4　　　a first circuit for passing said voice call data to the internet protocol network;
5　　　a second circuit for polling the internet protocol network about traffic information
6　transmitted therein; and
7　　　a third circuit for processing the polled information to determine whether the
8　voice call data is to be accepted by the internet protocol network.

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1 15. (original) The apparatus of claim 14 wherein said first circuit further comprises
2 one or more Ethernet cards that are connected to the internet protocol network.

1 16. (original) The apparatus of claim 14 wherein said second circuit is at least one
2 strongarm card.

1 17. (original) The apparatus of claim 16 wherein the strongarm card is connected to
2 the Ethernet card via a host CPU circuit.

1 18. (original) The apparatus of claim 14 wherein the third circuit compares a
2 parameter based on the polled information to a plurality of thresholds.

1 19. (original) The apparatus of claim 18 wherein the parameter is a packet loss ratio
2 defined as

3
$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})} \, .$$

1 20. (original) The apparatus of claim 19 wherein the third circuit compares the packet
2 loss ratio to a lower threshold and if the packet loss ratio is less than the lower threshold,
3 a new voice call is accepted into the internet protocol network.

1 21. (original) The apparatus of claim 19 wherein the third circuit compares the packet
2 loss ratio to the lower threshold and an upper threshold, and if lower threshold < packet
3 loss ratio < upper threshold, a new voice call is accepted into the internet protocol
4. network at a reduced bandwidth.

1 22. (original) The apparatus of claim 19 wherein the third circuit compares the packet
2 loss ratio to the upper threshold, and if the packet loss ratio is greater than the upper
3 threshold, a new voice call is blocked from entering the internet protocol network.

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PAGE 4/12 * RCVD AT 8/9/2007 11:41:54 AM [Eastern Daylight Time] * SVR:USPTO-EFXRF-3/6 * DNIS:2738300 * CSID:+17325309808 * DURATION (mm-ss):03-46